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APPLICANTS:	Klaus R. MOELLER et. al.	CONF. NO.:	1150
SERIAL NO.:	10/646,734	GROUP:	2614
FILED:	August 25, 2003	EXAMINER:	Lao, Lun-See
FOR:	NETWORKED SOUND MASKING AND PAGING SYSTEM		

APPELLANT'S BRIEF ON APPEAL UNDER 37 C.F.R. §41.37

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May 24, 2010

Dear Sir:

In accordance with the provisions of 37 C.F.R. §41.37, the Appellant submits the following:

I. 41.37(c)(1)(i) REAL PARTY IN INTEREST

The real party in interest is 777388 Ontario Limited.

II. 41.37(c)(1)(ii) RELATED APPEALS AND INTERFERENCES

None.

III. 41.37(c)(1)(iii) STATUS OF CLAIMS

Claims 108-119 are pending; with claims 108, 114 and 116 being written in independent form.[1]

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[1] See page 1 of the August 24, 2009 Final Office Action.

A. Claims 108-118 stand finally rejected under U.S.C. § 103(a) as being unpatentable over U.S. Patent No. 4,319,088 ("Orfield") in view of U.S. Patent Application Publication No. 2002/0072816 ("Shdema").[2]

B. Claim 119 stands finally rejected under U.S.C. § 103(a) as being unpatentable over U.S. Patent No. 4,319,088 ("Orfield") as modified by U.S. Patent Application Publication No. 2002/0072816 ("Shdema") as applied to claim 116, and in further view of U.S. Patent No. 4,686,693 ("Ritter").[3]

IV. 41.37(c)(1)(iv) STATUS OF AMENDMENTS

No amendments were filed after the Final Office Action mailed August 24, 2009.

[2] See page 2 of the August 24, 2009 Final Action.
[3] See page 10 of the August 24, 2009 Final Action.

V. 41.37(c)(1)(v) SUMMARY OF CLAIMED SUBJECT MATTER

A. Concise explanation of the subject matter set forth in each of independent claims 108, 114 and 116 and dependent claims 109-113, 115 and 117-119.

1. A general discussion of the subject matter of the application to assist the board of appeals in understanding example embodiments of the present application

Sound masking systems are widely used in offices and similar workplaces where an insufficient level of background sound results in diminished speech privacy. Such environments suffer from a high level of noise distractions, and lower comfort levels from an acoustic perspective. Sound masking systems operate on the principle of masking which involves generating a background sound in a given area. The background sound has the effect of limiting the ability to hear two sounds of similar sound pressure level and frequency simultaneously. By generating and distributing the background noise in the given area, the sound masking system masks or covers the propagation of other sounds in the area and thereby increases speech privacy, reduces the intrusion of unwanted noise, and improves the general acoustic comfort level in the area or space.[4]

Sound masking systems are of two main types: centrally deployed systems and independent self-contained systems. In a centrally deployed system, a central noise generating source supplies a series of loudspeakers installed throughout the physical area or space to be covered. The independent self-contained system comprises a number of individual self-contained sound masking units which are installed in the physical space. The sound masking units operate independently of each other, but may include a number of satellite speakers which extend the range of each self-contained, i.e. master, sound masking unit. Most sound masking systems include the capability for broadcast announcements and music over the loudspeakers contained in the sound masking units.[5]

The primary goal of sound masking systems is to provide an unobtrusive, effective masking sound that is adjustable for maximum consistency, and offers the

[4] See Page 1, lines 7-18 of the originally filed application.
[5] See Page 1, line 19 to Page 2, line 3 of the originally filed application

ability to meet the requirements of the occupants. The masking output is preferably sufficient to accommodate the existing acoustic requirements of the workplace environment and adjustable to handle changes to the acoustic characteristics of environment which occur over time. Similar demands are placed on the system for the public address and music functions. In short, the preferred sound masking system would produce an output with a frequency and volume level that is controllable to produce the desired acoustic response for workplace zones ranging in size from the smallest to larger spaces.[6]

Centralized systems are characterized by achieving uniformity of output, but not uniformity of acoustic response for the space. In a centralized system, the frequency spectrum of the sound masking output can only be adjusted at a centrally located equalizer, and as a result the sound masking output has the same frequency spectrum for all of the loudspeakers. Depending on the configuration of the centralized system, volume adjustments may be made for very large physical spaces, i.e. zones, by adjusting the amplifier output; for relatively smaller zones, volume adjustments are made by changing wiring connections or controls on the speaker enclosure, or by adjusting a hardwired zone volume control. In practice, it is difficult to accommodate environmental acoustic variations using a centralized system because the volume and frequency spectrum adjustments required for the typical physical zone size are too large to achieve a uniform acoustic result. A further disadvantage is that many of the adjustments for a centralized sound masking system require an installer or technician to re-enter the ceiling space or to rewire the speakers in the system.[7]

The independent self-contained system has a number of important advantages over the centralized arrangement. The independent self-contained system is more effective in terms of sound generation, volume adjustment, and frequency adjustment which, in turn, improves the performance of such systems as compared to centralized systems. In particular, the independent self-contained system provides a defined non-frequency specific output range for the masking output spectrum, and adjustments can be made at each master sound masking unit. The volume controls for an independent self-contained system also provide more flexibility than in the centralized

[6] See Page 2, lines 4-13 of the originally filed application.
[7] See Page 2, lines 14-29 of the originally filed application.

system, and provide for finer adjustments in smaller zones, in addition to centralized volume controls for large zone or global adjustment. However, with existing systems it is still necessary to re-enter the ceiling to adjust the frequency spectrum and volume output level for each master sound masking unit, and the controls for providing multi-unit volume zone adjustments require the hardwiring of the units.[8]

While existing independent self-contained systems are more flexible than centralized systems in many regards, they do not satisfy all the requirements of an ideal sound masking system as discussed above. Furthermore, other shortcomings are associated with existing sound masking systems. In both centralized and independent self-contained systems, the public address and music volume controls are limited in the same manner as described above for sound masking output volume controls. Second, any centrally located controls only affect the output level for the speakers or sound masking units which have a hardwired connection. It will be appreciated that this severely limits the adjustability of the system to future changes in the acoustic environment unless at least some of the system is rewired. Third, the tuning procedure for existing systems is time consuming and can still be inaccurate over the system even when undertaken with the appropriate level of skill and attention. And fourthly, adjustments to existing systems must be made on-site.[9]

Accordingly, there was a need for a networked sound masking system with individually controllable and programmable sound masking units, and which system is easily adaptable to changing sound qualities in a physical space or spaces in a building environment.[10]

As shown in FIG. 1, the networked sound masking system 10 comprises a control unit 12, and a network 11 comprising a plurality of standard master sound masking units or master hub units 14, indicated individually by 14a, 14b, 14c,...14n, one or more master sound masking switch units or master switch hubs 16, one or more master sound masking power units or master power hubs 18, and one or more satellite units or hubs 20, indicated individually by 20a, 20b, 20c. The physical

[8] See Page 3, lines 1-14 of the originally filed application.
[9] See Page 3, lines 15-28 of the originally filed application.
[10] See Page 3, line 29 to Page 4, line 2 of the originally filed application.

connections for the network 11 between the master sound masking units 14, 16, 18 may comprise 5 or 4 conductors. In a 5 conductor arrangement, two conductors carry power, two conductors provide a communication channel for control and paging signals, and one conductor provides for ground in an AC powered implementation. (In a DC implementation, the conductor for ground may be eliminated). The conductors are preferably terminated with multi-pin connectors as described below.[11]

The master hubs 14 serve as junction boxes in the network 11. The master hub switch 16 provides a connection to an "In-room" wall switch 24 located in a physical space, e.g. a room. The In-room wall switch 24 is coupled to the network 11 through the master switch hub 16 or either directly as described below with reference to FIG. 21. As will also be described in more detail below, the In-room switch 24 allows sound masking parameters to be adjusted or set locally. The In-room switch 24 may include an integral or separate In-room remote sensor module 26 to allow the settings to be adjusted using a remote control unit 28, for example, a handheld IR or wireless based unit. The master power hub 18 provides power for additional master hubs 14, 16 which are connected to the master power hub 18. As shown in FIG. 1, the master power hub 18 includes a power supply 30 for providing the additional power. The master hub 14d coupled to the master power hub 18 is supplied with additional power from the power supply 30. [12]

Referring to FIG. 1, the control unit 12 also includes a power supply unit 32, for example, a DC power supply, for providing a power feed to the units coupled to the network 11. The control unit 12 may also include a communication/control link 34 to a computer 36, for example, a personal computer or PC. Through software the computer 36 provides an interface for configuring, administering, and running diagnostics. The software on the computer 36 also provides the equalization or tuning function as described in more detail below. The communication interface 34 provides the capability to access the control unit 12 from a remote location, e.g. within the building or from an offsite location. The communication interface 34 may comprise a wireless link, a telephone communication, radio communication, computer network (e.g. a Local Area Network (LAN) or a Wide Area Network (WAN)), or a connection

[11] See Page 9, line 20 to Page 10, line 8 of the originally filed application.
[12] See Page 10, lines 9-22 of the originally filed application.

through the Internet or World Wide Web (WWW). This provides greater flexibility in configuring, adjusting and maintaining the sound masking system 10 from a remote or off-site location, for example, a wireless link or a Wide Area Network or Internet link.[13]

The computer 36 may be used for tuning the equalization function in the master sound masking hubs 14, 16, 18 as will be described in more detail below with reference to FIG. 14. For the tuning function, the computer 36 is equipped with appropriate software for performing the tuning functions and a microphone 38 or a sound level meter 39. The microphone 38 functions as a transducer to convert acoustical measurements into a form suitable for analysis by the software running on the computer 36. For the tuning function, the computer 36 preferably comprises a notebook computer with a wireless link for the communication link 34.[14]

As shown in FIG. 1, speakers 22, denoted individually by references 22a, 22b, 22c, 22d, 22e, 22f, 22g, 22h, 22i, . . . plug into the master hubs 14, the switch master hubs 16, the power master hubs 18 and the satellite hubs 20. The individual speakers 22 may comprise devices which are suspended above the ceiling tiles or a speaker integrated with ceiling plate adapter 23 which is mounted in the ceiling surface. It will be appreciated that other types of speaker enclosures and installations are also contemplated. For some installations, it may be advantageous to combine the master hub 14, 16, 18 and speaker 22 in a single or integrated enclosure.[15]

According to another aspect of the invention, additional control units, indicated individually by reference 13a to 13n, may be coupled to the control unit 12, for example, in a daisy chain. The control unit 13a is coupled to one or more master hubs 17, indicated individually by references 17a, 17b, . . . 17i, to form another network or zone 15a. As shown, a speaker 23, indicated individually by references 23a, 23b, . . . 23i is coupled to the respective master hub units 17. In addition to the master hub unit 17, the network 15 may include master switch hubs, master power hubs, and satellite hubs as described above. The control unit 13a and network 15a allow a

[13] See Page 10, line 23 to Page 11, line 8 of the originally filed application.
[14] See Page 11, lines 9-17 of the originally filed application.
[15] See Page 11, lines 18-26 of the originally filed application.

networked sound masking system to be configured for another physical space or zone in a building, e.g. another floor, while still be connected to the control unit 12 in order to provide a centralized control facility. Similarly, the nth control unit 13n is coupled to one or more master hub units 19a, 19b, including a master power unit 19i, and/or master switch unit (not shown) and satellite hubs (not shown). As shown, a speaker 25, indicated individually by references 25a, 25b, . . . 25j, is coupled to the respective master hub units.[16]

The master sound masking hubs 14a, master switch hubs 16 and master power hubs 18 and the satellite sound masking hubs 20 together with the speakers 22 provide the sound masking functionality, i.e. sound masking signal generation and amplification. Each master sound masking hub 14, 16, 18 (and satellite sound masking hub 20) is configured either individually or as a group for a particular physical space, e.g. office, room, zone in a open office, etc. The master sound masking hubs 14, 16, 18 are configured to generate a specific sound masking signal at a specified output level for performing the sound masking in the physical space. As will be described in more detail below, the sound masking signal is generated according to programmable spectrum, equalizer, and volume settings. The satellite sound masking units 20 are connected to their associated master unit 14 (16 or 18) and reproduce the sound masking signal generated by the master unit 14. The satellite units 20 provide a cost-effective way to expand the coverage of the master sound masking unit 14 (16 or 18) in a building space.[17]

The control unit 12, as will be described in more detail, couples to the network 11 and provides the capability to adjust the functional aspects of the master sound masking hubs 14, 16, 18 and the satellite sound masking hubs 20. The sound masking functions include masking signal spectrum, masking signal output volume, and paging volume. The control unit 12 also provides diagnostic functions and timer control functions.[18]

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[16] See Page 11, line 27 to Page 12, line 13 of the originally filed application.
[17] See Page 12, lines 14-28 of the originally filed application.
[18] See Page 12, line 29 to Page 13, line 4 of the originally filed application.

The control unit 12 configures the network 11 by assigning identities or addresses to each of the master hubs 14, 16, 18. The addressing of the individual master hubs 14, 16, 18 enables the control unit 12 to direct commands and/or status requests to individual master sound masking hubs 14, 16, 18 (and indirectly the associated satellite sound masking hubs 20, i.e. via the master hubs 14, 16, 18), or to groups of master sound masking hubs 14, 16, 18, or to the entire network 11 as a whole. The control unit 12 is then used to set/adjust the masking signal spectrum, the masking signal volume, and/or the paging volume for the selected (i.e. addressed) master hub 14, 16, 18 and the satellite sound masking hub 20. According to another aspect, the master sound masking hubs 14, 16, 18 includes a digital equalizer for providing greater programming flexibility over the spectrum for the sound masking signal generated by the selected master and satellite sound masking hubs 14, 16 or 18 and 20. The master hubs 14, 16, 18 may include another digital equalizer for the paging signal.[19]

Reference is next made to FIG. 2 which shows the master sound masking hub 14 in greater detail. As shown, the master unit 14 comprises a digital signal processing module 50, an audio power amplifier stage 52, an input serial interface 54, an output serial interface 56, and a power supply module 58. The input serial interface 54 and the output serial interface 56 form a communication interface which provides the capability to communicate with the control unit 12 and other master sound masking hubs 14, 16 or 18, and/or the In-room wall switch 24 connected in the network 11. The master hub 14 includes a local power supply 58 for powering the circuitry. The audio power amplifier stage 52 drives the speaker 22 (FIG. 1) which emits the sound masking signal or a paging signal as will be described in more detail below. The audio power amplifier stage 52 also includes an output port 60 for coupling to a satellite hub 20 (FIG. 1).[20]

The master switch hub 16 (FIG. 1) and the master power hub 18 have essentially the same topology as the master hub 14 depicted in FIG. 2. The master switch hub 16 includes a connection for the In-room wall switch 24 for coupling to the network 11. (Alternatively, the In-room wall switch 24 is connected directly to the

[19] See Page 13, lines 5-19 of the originally filed application.
[20] See Page 13 line 20 to Page 14, line 2 of the originally filed application.

network 11). The master power hub 18 (FIG. 1) includes a power input 64. The power input 64 receives a power feed from the auxiliary DC/AC power supply 30 described above with reference to FIG. 1.[21]

As shown in FIG. 2, the digital signal processing module 50 is implemented as a single chip DSP device such as the MC56F801 available from the Motorola Corporation. The digital signal processing module 50 comprises a random noise generator module 66, an equalizer module for sound masking 68, an equalizer module for paging 69, a pulse width modulator or PWM stage 70, a switching logic stage 72, and a paging demultiplexer module 74. The digital signal processing module 50 includes a processing unit 76 (i.e. a microprocessor) in addition to on-chip resources such as a memory. The processing unit 76 controls the operation of the modules 66, 68, 69, 70, 72 and 74 to provide the functionality as described in more detail below.[22]

The random noise generator module 66 is the signal source for generating the sound masking signal. According to this aspect, the random noise generator module 66 is implemented as a firmware module executed by the processing unit 76. The equalizer 68 for the masking signal, the equalizer 69 for the paging signal, and the paging demultiplexer 74 are implemented in firmware as functions executed by the processing unit 76. The random noise generator may comprise a multi-stage shift register and an Exclusive-OR gate network operating on one register output or two register outputs as described in U.S. Pat. No. 4,914,706, which issued to the applicant in common with the subject application and is incorporated herein by reference.[23]

The equalizer module 68 comprises a 1/3 Octave equalizer which is used for adjusting the sound spectrum of the noise signal output to the desired contour. The equalizer module 68 for the sound masking signal provides twenty-three (23) bands. In the present embodiment, the 1/3 Octave Band frequencies comprise 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, and 10,000 Hertz (Hz). The output from the equalizer module 68 is

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[21] See Page 14, lines 3-9 of the originally filed application.
[22] See Page 14, lines 10-19 of the originally filed application.
[23] See Page 14, lines 20-29 of the originally filed application.

a sound masking signal with a controllable contour which is coupled through the PWM module 70 to the amplifier power stage 52.[24]

The PWM module 70 input to the amplifier power stage 52 functions to convert the digitally generated sound masking signal into an analog signal equivalent. It will be appreciated that in this context the output from the PWM module 70 functions under firmware control as a digital-to-analog converter or DAC.[25]

The audio power stage 52 provides an amplified output level for the contoured sound masking signal. The contoured sound masked signal is amplified by the audio power stage 52 and output to the connected speaker 22 which emits a sound masking sound into the physical space. The audio power stage 52 also provides an amplified output for the paging signal which may comprise announcements, emergency notifications, background music or other broadcasts over the speaker. In the preferred embodiment, the output level of the audio power stage 52 is controllable by the processing unit 76 through the pulse width modulation of the input signal which is fed to the audio power stage 52.[26]

The equalizer module 69 for the paging signal is implemented in a similar fashion but with a fewer number of bands.[27]

Referring to FIG. 2, the switching logic stage 72 together with the input serial interface stage 54 and the output serial interface stage 56 form a communication interface, indicated by reference 55 for the master hub 14. The communication interface 55 couples the processing unit 76 in the DSP 50 to the network 11 (FIG. 1) and allows the master hub 14 to receive control commands and transmit responses. Paging signals/audio data sent by the control unit 12 over the network 11 are also received via the communication interface 55. The switching logic stage 72 connects the processing unit 76 to the input and output serial interface stages 54 and 56. The input serial interface 54 allows the processing unit 76 to communicate with an

[24] See Page 15, lines 1-9 of the originally filed application.
[25] See Page 15, lines 10-14 of the originally filed application.
[26] See Page 15, lines 15-24 of the originally filed application.
[27] See Page 15, lines 25-26 of the originally filed application.

upstream device, for example, the master sound masking unit 14a (FIG. 1) or the control unit 12 (FIG. 1). The output serial interface stage 56 allows the processing unit 76 to communicate with a downstream device, for example, the master sound masking switch hub 16 (FIG. 1). In conjunction with the switching logic stage 72, the processing unit 76 monitors the serially encoded messages and acts upon messages which are addressed to the specified master sound masking hub 14. Each of the master sound masking hubs 14, 16 and 18 is assigned an address according to a self-addressing mechanism as will be described in more detail below.[28]

The satellite sound masking hubs 20 (FIG. 1) are associated with respective master sound masking hubs 14, 16 or 18. The satellite sound masking hubs 20 are connected to a speaker 22, and are coupled to one of the master sound masking hubs 14, 16 or 18. The satellite sound masking hubs 20 act as slaves or satellites to the master sound masking hub 14 (16 or 18) and reproduce the sound masking signal output generated by the associated master sound masking hub 14 (16 or 18).[29]

Reference is next made to FIG. 3(a), which shows the control unit 12 in more detail. As shown, the control unit 12 comprises a processor unit (i.e. a microprocessor) 80, a program memory 82, a data memory 84, a display module 86, a keypad 88, a real-time clock module 90, a parameter memory 92, a first serial communication interface 94, a communication interface 96, and a second serial communication interface 98. The first serial communication interface 94 couples the control unit 12 to the master sound masking hub 14, 16, or 18 in the network 11 (FIG. 1). The second serial communication interface 98 provides a communication interface for coupling the control unit 12 to the other control unit 12a (FIG. 1). The communication interface 96 provides the communication link to the computer or PC 34 as described above with reference to FIG. 1.[30]

Reference is next made to FIG. 4, which shows functional modules in the control unit 12 for performing various functions associated with the networked sound masking system 10. The control unit 12 includes a functional module 100 for

[28] See Page 15, line 27 to Page 16, line 16 of the originally filed application.
[29] See Page 16, lines 17-21 of the originally filed application.
[30] See Page 16, line 22 to Page 17, line 4 of the originally filed application.

providing time of day and date functions, a functional module 102 to control an 1/n Octave equalizer for the paging feature, a functional module 104 for providing paging zones and selecting the paging signal for the paging zones, a functional module 106 for adjusting the frequency spectrum of the contoured sound masking signal output for one or more of the hubs 14, 16 or 18 according to preset equalization functions, a functional module 108 to provide timer functions for the system 10, a functional module 109 to provide timer zone/schedule set-up functions, a functional module 110 to control communication functions with the computer 36 (FIG. 1), the master hubs 14, 16, 18 (FIG. 1), and the In-room switch 24 (FIG. 1), a functional module 112 to provide volume control for the sound masking signal output, a functional module 114 to provide paging volume control, a functional module 116 to provide system configuration functions (including self-addressing, i.e. the addressing of the master sound masking hubs 14, 16, 18 in the network 11), a functional module 118 for locating particular hubs or units in the network, a functional module 120 for performing diagnostic functions, and a functional module 122 for processing inputs from the In-room switch 24 (FIG. 1). The operation of the functional modules in the control unit 12 is now described in more detail with reference to the flowcharts in FIGS. 5 to 19.[31]

Reference is first made to FIG. 5, which shows a start-up process 200 for the control unit 12. The start-up process 200 is executed in response to a power-up 202 or a reset condition. The start-up process 200 comprises an initialization step 204 which includes configuring the control unit 12. After the initialization step 204, the control unit 12 runs a timer operation 206, a switch operation 208, and a display/setup operation 210. The display/setup operation 210 is executed as a background task, for example, in a polling loop. The timer operation 206 is periodically executed, for example, on an interrupt driven basis or as part of a polling loop in the display/setup operation. The switch operation 208, i.e. sensing inputs from the In-room switch 24 (FIG. 1) is also periodically executed.[32]

Reference is next made to FIG. 6, which shows the display/setup operation 210 in more detail. As shown the display/setup operation 210 comprises displaying a

[31] See Page 17, line 29 to Page 18, line 20 of the originally filed application.
[32] See Page 18, lines 21-30 of the originally filed application.

series of menu functions on the display 86 (FIG. 3) which are accessed via selections from the keypad 88 (FIG. 3). As shown in FIG. 6, the menu functions include a Date/Time function menu 212, a Volume function menu 214, a Contour menu function 216, a Unit (i.e. hub) Locator function menu 218, an Equalizer Setting function menu 220, a Switch function menu 222, a Paging Zone/input function 224, a Timer Zone/Schedule function menu 226, a Diagnostic function menu 228, a System Configuration function menu 230, and Serial Number function menu 232. According to another aspect, the functionality of the control unit 12 may be implemented in the computer 36.[33]

The processing steps for the Date/Time function menu 212 are shown in FIG. 7. The first step comprises displaying the time of day 240 and prompting the user to change the time of day 242. If the user selects to change the time of day, then a set time procedure 244 is executed. Otherwise the date is displayed 246, and the user is prompted to change the date 248. If the user selects to change the date, a set date procedure 250 is executed.[34]

Reference is next made to FIG. 8, which shows in more detail the processing steps for setting the Masking and Paging volume function 214, the Contour Control function for the sound masking signal 216, and the Equalizer Band function 220 for the paging signal. The steps for controlling each of these functions is implemented according to a process 213 as illustrated in FIG. 8. As shown, the first step in block 260 comprises selecting the paging function or the sound masking function. The next step in block 262 comprises selecting the master sound masking hub. In decision block 264, a selection is made between a single master sound masking hub 14, 16, 18 or multiple master sound masking hubs 14, 16 and/or 18. If multiple master sound masking hubs 14, 16, 18 are to be configured, then the next step 266 involves selecting the range for the sound masking hubs 14, 16, 18. The level for the sound masking hubs 14, 16, 18 in the range is entered in block 268 and transmitted via the network 11 to all the sound masking hubs 14, 16, 18 in the selected range. After the level has been sent to the sound masking hubs 14, 16, 18 in the range, in block 270,

[33] See Page 19, lines 1-11 of the originally filed application.
[34] See Page 19, lines 12-17 of the originally filed application.

and the first sound masking unit 14a in the range is selected, i.e. addressed, in block 272.[35]

Referring still to FIG. 8, the next step in block 274 involves reading the level setting for the master sound masking hub 14, 16 or 18 which was selected in step 272 or as a result of the branch from decision block 264. The level setting received from the sound masking hub 14, 16 or 18 is compared to the desired setting stored in the control unit 12, and if a change in the level is needed as determined in decision block 276, then the desired level setting is sent to the selected master sound masking hub 14, 16 or 18 in block 278. If no change is indicated for the selected master sound masking hub 14, 16 or 18', then the next master sound masking hub 14, 16 or 18 in the network 11 is selected, i.e. addressed, in block 280 and the steps 274 and 276 are repeated. The same processing steps are utilized for setting the Masking signal and Paging signal Volume function, the Contour Control function 216, and the Equalizer Band functions for the paging signals 220.[36]

Reference is next made to FIG. 9, which shows in more detail the processing steps for the Timer Zone/Schedule function menu 226. The first step in block 290 comprises setting timer zones in the network 11. The next step involves selecting the zone in block 292. Next in block 294, timer schedules, timer profiles, ramp-up schedules and exception dates are displayed for the selected zone, and the user is prompted to enter a change in the timer settings. If a change is entered (decision block 296), then the entry is stored in memory as indicated in block 298. The timer zones are independent from the switch and paging zones. The timer schedules may include pre-programmed profiles, such as, standard office settings, regular office hours, and executive office settings for ramp-up, timer schedules and exception dates. The ramp-up feature provides the capability to set timed schedules for ramping up the sound masking output level. Exception dates are programmed for dates such as holidays, and override the regular timer schedule.[37]

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[35] See Page 19, line 18 to Page 20, line 3 of the originally filed application.
[36] See Page 20, lines 4-16 of the originally filed application.
[37] See Page 20, lines 17-30 of the originally filed application.

Reference is next made to FIG. 10, which shows the operation of the diagnostic menu 228 and function 120 (FIG. 4) for the control unit 12 in more detail. The first step 300 involves the control unit 12 selecting the first of the master sound masking hubs 14, 16 or 18 for the diagnostic test. In response, the control unit 12 retrieves the serial number from the master sound masking hub 14, 16, 18 over the network 11 as indicated by block 302. If there is an error (as indicated by decision block 304), then a communication error (in block 306) is logged for that hub 14, 16 or 18 and another hub 14, 16 or 18 is selected in block 320. If there is no communication error (decision block 304), then the control unit 12 checks the serial number against the entry stored in a lookup table in block 308. If the serial number does not match the entry in the lookup table, then an identification error is logged in block 312, and another master sound masking hub 14, 16 or 18 is selected in block 320. If the serial number matches the entry in the lookup table (decision block 310), then the status for the master sound masking hub 14, 16 or 18 is queried by the control unit 12 in block 314. The status of the selected master sound masking hub 14, 16 or 18 is checked in block 316, and if the status is fail or does not meet specifications, then a status error is logged in block 318. The next step in block 320 involves selecting another master sound masking hub 14, 16 or 18 and repeating steps 302 through 320, as described above, until all, or the selected group, of the master sound masking hubs 14, 16 or 18 have been queried as determined in block 322. The last step in the operation of the diagnostic function module 120 comprises generating and/or displaying a diagnostics report as indicated in block 324.[38]

Reference is next made to FIG. 11(a), which shows the operation of the system configuration and self addressing functional module 116 and menu function 230 for the control unit 12 in more detail. The control unit 12 is preferably password protected, and the first step 330 involves prompting the user to enter a password. If the password is incorrect (decision block 332), then further access is denied (block 334). If the entered password is correct, the password is displayed in block 336, and the user is given the option of changing the password (decision block 338). If the user changes the password, then the new password is saved in block 340. The next step 342 involves displaying the number of master sound masking hubs 14, 16, 18 that are

[38] See Page 21, lines 1-24 of the originally filed application.

presently configured for the network 11. If the system 10 is being setup for the first time, the number of hubs or units may be configured at the factory or entered in the field by the technician. The user is given the option of changing the number of hubs 14, 16, 18 configured for the system 10 in decision block 344, and the new number of hubs 14, 16, 18 is stored in step 346.[39]

Referring still to FIG. 11(a), in decision block 348, the user is prompted to initialize the system 10. If the user elects to initialize the system 10, then the control unit 12 executes an initialization procedure indicated generally by reference 350. The initialization procedure 350 is shown in more detail in FIG. 11(b). As shown, the first step 352 in the initialization procedure 350 involves resetting all of the master sound masking hubs 14, 16, 18 connected to the network 11. As a result of the reset operation 352, each of the master sound masking hubs 14, 16, 18 has a logical address of 0. Since all of the hubs 14, 16, 18 have logical address 0, the first sound masking hub 14, 16 or 18, i.e. the master hub 14a, responds when the control unit 12 queries the hubs 14, 16, 18 as indicated by block 354. The selected hub 14, 16, 18 is then queried for its serial number in block 356. The serial number is assigned to the hub 14, 16, 18 at the time of manufacture and preferably comprises a code stored in non-volatile memory in the hub 14, 16, 18. The control unit 12 uses the serial number to generate a unit address, i.e. logical address, for the hub 14, 16, 18 as indicated in block 358. The serial number is preferably stored in memory, for example a look-up table in the control unit 12, and provides a cross-reference to the master sound masking hub 14, 16, 18. The current logical address generated in step 358 is unique for the master hub 14, 16, 18 in the present network configuration 10, but for another network configuration the logical addresses may be regenerated. Following the addressing operation, the next sound masking hub 14 is selected by the control unit 12 and the current logical address is incremented for the next sound masking hub 14, 16, or 18. The operations for assigning the current logical address to the master sound masking hub 14, 16, 18 based on the serial number according to steps 356 to 360 are repeated. These operations are repeated until all of the sound masking hubs 14, 16, 18 have been assigned current logical addresses by the control unit 12 as indicated by decision block 362. Following this scheme, the current logical address for the last

[39] See Page 21, line 25 to Page 22, line 8 of the originally filed application.

sound masking hub 14, 16, 18 is equal to the number of sound masking hubs 14, 16, 18 connected to the networked system 10.[40]

Reference is next made to FIG. 12, which shows the timer function 206 (FIG. 5) in more detail. In response to an interrupt or a request from a polling loop, a wake-up call or "clock tick" is periodically issued as indicated in step 370, and a schedule of timed events is checked in block 372. The timed events may comprise, for example, changes in the level of the sound masking signal for all or some of the master sound masking hubs 14, 16, 18 (and the associated satellite sound masking hubs 20). If the schedule indicates that there is no change in sound masking level, then the timer function 206 goes to sleep (block 376). If there is a scheduled change, then the new level for the sound masking signal is transmitted via the network 11 to the affected sound masking hubs 14, 16, 18 (block 378).[41]

Reference is next made to FIG. 13, which shows in flowchart form a method for selecting control functions in the control unit 12 for controlling the master sound masking units 14. As shown, the control functions 400 include an initialization procedure 401, a program serial number procedure 402, a read serial number procedure 403, an assign logical address procedure 404, a read level procedure 405, and a write level procedure 406.[42]

The initialization procedure 401 comprises a function 408 for resetting the logical addresses and a function 410 for writing logical addresses for the master sound masking hubs 14, 16, 18 as described above with reference to FIG. 11. The program serial number procedure 402 provides a mechanism for programming or regenerating the serial number stored in non-volatile memory for each hub 14, 16, 18. The procedure 402 comprises a write serial number function 412. The read serial number procedure 403 comprises a read serial number function 414 which the control unit 12 utilizes to read the serial numbers of the hubs 14, 16, 18, for example, as described above with reference to FIG. 11. The assign logical address procedure 404 comprises a

[40] See Page 22, line 9 to Page 23, line 7 of the originally filed application.
[41] See Page 23, lines 8-18 of the originally filed application.
[42] See Page 23, lines 19-24 of the originally filed application.

write address function 416 for writing, i.e. assigning, logical addresses to the sound masking hubs 14, 16, 18. The read level procedure 405 comprises a read level function 418 which allows the control unit 12 to read the current levels for the various settings for the hubs 14, 16, 18 being addressed by the control unit 12 or by an In-room switch. The write level procedure 406 comprises a write level function 420 which allows the control unit 12 to write the level for the selected function for the sound masking signal in the master sound masking hub 14, 16, 18 being addressed by the control unit 12. Once the master sound masking hub 14, 16, 18 is selected, the control unit 12 next selects the function to be queried/programmed, and then reads the parameter setting using the read level function 418, or writes the parameter setting, using the write level function 420.[43]

As described above, the master sound masking hubs 14, 16, 18 according to the present invention include an equalizer stage 68 (FIG. 2) which allows the shaping of the sound spectrum of the sound masking noise signal output. In addition, the hubs 14, 16, 18 also includes the second equalizer stage 69 (FIG. 2) to allow for shaping the spectral characteristics of the paging signal. Advantageously, the capability to address each of the sound masking hubs allows the equalizer stages 68, 69 to be individually set for each of the hubs 14, 16, 18 or a group of the hubs 14, 16, 18, and this capability greatly enhances the functionality of the networked sound masking system 10 according to the present invention.[44]

Reference is made to FIG. 14 which shows a procedure 430 according to another aspect of the invention for controlling the equalizer function in each of the sound masking hubs 14, 16, 18. According to this aspect, the equalizer functions are performed in the computer 36. The computer 36 and the microphone 38 are used to take sound level readings for the physical space. Calculated control settings based on these readings are transmitted by the computer 36 via the communication link 34, e.g. wireless link, to the control unit 12, which then transmits control data to the hubs 14, 16, 18 affected. As will now be described with reference to FIG. 14, the readings from the microphone 38 or the sound level meter 39 are used in conjunction with settings in a Prescribed Contour Table stored in the computer 36 to adjust the level settings in

[43] See Page 23, line 25 to Page 24, line 16 of the originally filed application.
[44] See Page 24, lines 17-26 of the originally filed application.

the equalizer stages 68 for the master sound masking hubs 14, 16, 18. It will be appreciated that the Prescribed Contour Table defines the ideal sound masking levels for the physical space, and the levels are programmable or user-definable.[45]

As shown in FIG. 14, the first operation in the equalization procedure 430 comprises receiving the unit ID(s) (entered by a user or technician) to select the sound masking hub or hubs 14, 16, 18 on which the equalizer function is to be adjusted/programmed (block 431). The next step in block 432 involves selecting the equalization adjustment or tuning mode. If auto tuning mode is selected, then the next step in block 434 involves reading (and displaying) the current sound levels. Next in block 436, the sound levels are compared to prescribed settings stored in memory. The prescribed levels are user definable and may be determined, for example, by identifying acceptable sound level readings in decibels (dB) by band, with one band for every 1/n octave in the equalizer. A prescribed setting may comprise, for example, a 63 Hz band center at 46 dB+/2 dB. If the measured sound levels are within an acceptable range of the prescribed settings, then the auto-tuning procedure is concluded. If not within an acceptable range, then the equalization levels are modified by the computer 36 and applied to the relevant sound masking hubs 14, 16, 18 via the control unit 12, as indicated in block 438. Measurements for the modified levels set in block 438 are then taken as indicated in block 440, and these measurements are again compared to the prescribed settings as indicated in block 442. If the modified levels are within an acceptable range, then the auto-tuning procedure is concluded. If the measurements corresponding to the modified levels are not within the acceptable range as determined in block 442, then the required equalizer settings are calculated in block 443 and they are compared to the equalizer setting limits in block 444. The setting limits define maximum or minimum equalization settings, for example, zero (0) as the minimum and one hundred (100) as the maximum. As indicated, a comparison is made to determine if the required equalizer settings are "below minimum", "above maximum", or "within limits". If the required equalizer settings are within limits, then steps 438 to 442 are repeated. If the required equalizer settings are below minimum, then the frequency band(s) corresponding to those levels

[45] See Page 24, line 27 to Page 25, line 10 of the originally filed application.

are eliminated. If the required equalizer settings are above maximum, then the equalizer settings are set to maximum in block 448.[46]

Referring again to FIG. 14, in manual mode, the first step in block 450 involves taking sound level measurement and displaying the levels associated with those measurements. Next a decision is made to change the equalizer settings or to keep them the same in block 452, and if necessary the equalizer settings are changed in block 454. The process may then be repeated in step 450.[47]

As described above, the computer 36 and the microphone 38 or the sound level meter 39 provide an effective mechanism for adjusting the equalizer function in each of the sound masking hubs 14, 16, 18 through the control unit 12 and networked connection without the need for opening the ceiling tile to physically access any of the master sound masking units 14, 16, 18.[48]

As shown in FIG. 1, the In-room wall switch 24 is provided in a physical space, e.g. meeting room, and is connected to the master switch hub 16 or alternatively the In-room wall switch 24 is coupled directly to the network 11. The In-room wall switch 24 provides the capability for an occupant to manually adjust the output characteristics of the master hubs 14, 16 or 18 (and the associated speakers 22) configured to be associated with the In-room wall switch 24. The In-room wall switch 24 may include the In-room remote sensor 26 for use with the In-room remote control 28, for example, a handheld wireless IR device. The In-room wall switch 24 may-be implemented as depicted in FIG. 21.[49]

As shown in FIG. 21, the In-room wall switch 24 comprises a switch panel 470, a display 472, a processing unit 474, and a communication interface 476. The communication interface 476 couples the In-room wall switch 24 to the master sound masking switch hub 16 or directly to the network 11. The communication interface 476 comprises a first serial interface module 478, a switching logic stage 480, and a second serial interface module 482. The processing unit 474 uses the switching logic

[46] See Page 25, line 11 to Page 26, line 10 of the originally filed application.
[47] See Page 26, lines 11-16 of the originally filed application.
[48] See Page 26, lines 17-21 of the originally filed application.
[49] See Page 26, line 22 to Page 27, line 2 of the originally filed application.

stage 480 to send control messages and receive display messages from the master switch hub 16 or the control unit 12 via the network 11. The processing unit 474 uses the display 472 to display status and operating information, typically received from the control unit 12. As shown, the switch panel 470 comprises a paging/sound masking function select button 484, an adjust up button 486, an adjust down button 488, and a mute button 490. Depressing the paging/sound masking button 484 to select the sound masking adjust function causes a sound masking LED 485 to be activated, and depressing the up button 486 increases the output level of the contoured sound masking output signal, while depressing the down button 488 decreases the output level of the output signal. If the paging function is selected using the button 484, a paging LED 487 is activated, and depressing the up 486 and the down 488 buttons increases or decreases the volume level of the paging signal. The mute button 490 allows the selected signal output, i.e. sound masking or paging, to be muted.[50]

According to another aspect, the In-room wall switch 24 may be provided with an interface 489 for receiving control signals from the In-room remote sensor 26 and the wireless remote 28. The wireless remote 28 provides the functionality of the switch panel 470, i.e. sound masking/paging select, up and down adjust, mute, in a portable handheld unit.[51]

Reference is next made to FIG. 15, which shows the operation of the switching function module 122 and the switch menu 22 for the control unit 12. The first step 500 as shown in FIG. 15 comprises selecting the first In-room switch 26. The next step 502 involves assigning one or more of the master hubs 14, 16 or 18 to the selected switch 24. The process is repeated for the next In-room switch 26 as indicated in block 504.[52]

Reference is next made to FIG. 16, which shows the primary operations performed by the processing unit 474 in the In-room wall switch 24 (FIG. 21). After power-up (block 510), the processing unit 474 scans the select button 484 to

[50] See Page 27, lines 3-23 of the originally filed application.
[51] See Page 27, lines 24-28 of the originally filed application.
[52] See Page 28, lines 1-6 of the originally filed application.

determine if the sound masking or paging function has been selected in block 512. Next in block 514, the parameters associated with the selected function are read, and then sent to the control unit 12 (i.e. via the response channel 152 (FIG. 3(b)). The control unit 12 then executes the change for the hubs 14, 16, 18 or 20 associated with that In-room switch 24.[53]

Reference is next made to FIG. 17 which shows the processing steps executed by the control unit 12 for the operation of the volume setting inputs from the In-room wall switch 26. The first step performed by the control unit 12 in block 520 involves selecting the first In-room wall switch 26 via the master switch hub 16 which is coupled to the In-room switch 26 through the communication interface 476 (FIG. 21). Once the In-room switch 26 is selected, the control unit 12 determines the sound masking signal output level (and the paging output volume level) from values stored in memory (block 522). Next the control unit 12 determines if there is a change in the sound masking (or paging output) volume level in block 524. As described above with reference to FIG. 21, a change in volume level is initiated by selecting the sound masking function or the paging volume function, and then depressing continuously or repeatedly the up or down button. In response, the control unit 12 sends a control message to the master sound masking hubs 14, 16, 18 programmed or associated with the In-room wall switch 24. The control message corresponds to the level setting as determined from the In-room wall switch 24. If no change is indicated for the In-room wall switch 24 in block 534, then the next In-room wall switch 24 in the network 11 is selected and the processing steps are repeated as described above.[54]

[0088] Reference is next made to FIG. 18 the processing steps for the unit locator function 118 (FIG. 4) and the locator menu function 218 (FIG. 6) in the control unit 12. The first step indicated in block 540 involves selecting the master sound masking hub 14, 16 or 18 in the network 11. Once selected, the control unit 12 sends a locator message or signal to the selected hub 14, 16 or 18 in step 542.[55]

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[53] See Page 28, lines 7-14 of the originally filed application.
[54] See Page 28, line 15 to Page 29, line 3 of the originally filed application.
[55] See Page 29, lines 4-8 of the originally filed application.

2. The following explains the subject matter set forth in independent claims 108, 114 and 116 and dependent claims 109-113, 115 and 117-119, referring to the specification by page and line number, and to the drawings, if any, by reference characters in accordance with 37 C.F.R. §41.37(c)(1)(v)

Claim 108

Claim 108 is directed to a sound masking system for masking sound in a physical environment. The system according to claim 108 includes a communication network for the physical environment. This reads on, for example, network 11 in Fig. 1.[56] The system of claim 108 further includes a plurality of sound masking units. At least some of the sound masking units include a digital processor configured for a sound masking signal generator and a communication interface, which is for coupling to the communication network and for receiving a plurality of control signals over the communication network. The control signals include a masking volume signal and a masking frequency signal. The sound masking signal generator is responsive to the masking volume signal and the sound masking frequency signal for generating a sound masking output signal. The sound masking output signal has a volume derived from the sound masking output signal and a frequency characteristic derived from the sound masking frequency signal. This reads on, for example, master sound masking units[57] 14 which include the digital signal processing module[58] 50 and the communication interface[59] 55. This system of claim 108 further includes a control unit configured to generate the control signals including the masking volume signal and the masking frequency signal, and the control unit includes a communication interface for coupling to the communication network for transmitting the control signals to selectively control operation of the plurality of sound masking units. This reads on, for example, the control unit 12.[60]

Claims 109-113

Claim 109 is dependent from claim 108, and includes all the elements of claim 108. Claim 109 further defines the sound masking unit as including an address component for recognizing control signals intended for the sound masking unit

[56] See element 11 in Fig. 1 and Page 9, line 20 to Page 10, line 8 of the originally filed application.
[57] See element 14 in Fig. 1 and Page 13, line 20 to Page 14, line 2 of the originally filed application.
[58] See element 50 in Fig. 2 and Page 14, lines 10-19 of the originally filed application.
[59] See element 55 in Fig. 2 and Page 15, line 27 to Page 16, line 16 of the originally filed application.
[60] See element 12 in Figs. 1 and 3, and Page 12, line 29 to Page 13, line 19.

associated with said address component. This reads on, for example, the combination of the digital signal processing module 50 and executable code or firmware.[61]

Claim 110 is dependent from claim 108, and includes all the elements of claim 108. Claim 110 further defines the plurality of sound masking units as being associated with a plurality of sound masking zones, and each of the sound masking units being associated with one of the plurality of sound masking zones, and the sound masking units providing sound masking for the associated sound masking zone independently of the other sound masking zones. This reads on, for example, the capability to control master sound masking units individually and/or in groups.[62]

Claim 111 is dependent from claim 110 and includes all of the elements of claims 110 and claim 108. Claim 111 further defines the sound masking units associated with each of said sound masking zones as being configured to provide a sound masking output tailored for the associated sound masking zone and the sound masking output being based on the masking volume and the masking frequency signals. This reads on, for example, the control unit 12.[63]

Claim 112 is dependent from claim 108 and includes all of the limitations of claim 108. Claim 112 further defines the sound masking system as comprising a plurality of zones, and one or more of the sound masking units being configured for one or more of the zones. This reads on, for example, the master sound masking units 14 and the control unit 12.[64]

Claim 113 is dependent from claim 112, and includes all of the limitations of claim 112 and claim 108. Claim 113 further defines the zones as including one or more of a sound masking zone, a timer zone, and a switch zone. This reads on, for example, the master sound masking units 14, the In-room switch 24, and the control unit 12.[65]

[61] See elements 50, 55 in Fig. 2, and Page 15, line 27 to Page 16, line 16.
[62] See elements 11, 12, 14, 16 and 18 in Fig. 1, and Page 13, lines 5-17, Page 19, line 18 to Page 20, line 16, and Page 24, lines 17-26.
[63] See element 12 in Figs. 1 and 3, and Page 12, line 29 to Page 14, line 2.
[64] See elements 12 and 14 in Figs. 1 and 2, Page 12, line 14 to Page 13, line 19.
[65] See elements 12, 14 and 24, Page 14 to Page 13, line 19 and Page 28, line 7 to Page 29, line 3 of the originally filed application.

Claim 114

Claim 114 is directed to a sound masking system for controlling the ambient noise in a physical environment. The system of claim 114 includes a communication network in the physical environment. This reads on, for example, network 11 in Fig. 1.[66] The system of claim 114 further includes a plurality of sound masking units. At least some of the sound masking units include a sound masking generator, which includes a digital processor and a communication interface. The communication interface is configured to couple to the communication network and to receive a plurality of control signals over the communication network. The control signals include a masking volume signal and a masking frequency signal. The sound masking signal generator is responsive to the masking volume signal and the sound masking frequency signal for generating a sound masking output signal. This reads on, for example, master sound masking units[67] 14 which include the digital signal processing module[68] 50 and the communication interface[69] 55. This system of claim 114 includes a control unit configured to generate the control signals including the masking volume signal and the masking frequency signal, and the control unit includes a communication interface for coupling to the communication network for transmitting the control signals to selectively control operation of the plurality of sound masking units. This reads on, for example, the control unit 12.[70] The system of claim 114 further includes a plurality of zones, with one or more of the sounds masking units being configured for one or more of the plurality of zones. This reads on, for example, the master sound masking units 14 and the control unit 12.[71]

Claim 115

Claim 115 is dependent from claim 114, and includes all the elements of claim 114. Claim 115 further defines further defines the zones as including one or more of a sound masking zone, a non-masking zone, a timer zone, and a switch zone. This reads

[66] See element 11 in Fig. 1, and Page 8, line 20 to Page 9, line 8 of the originally filed application.
[67] See element 14 in Fig. 1 and Page 13, line 20 to Page 14, line 2 of the originally filed application.
[68] See element 50 in Fig. 2 and Page 14, lines 10-19 of the originally filed application.
[69] See element 55 in Fig. 2 and Page 15, line 27 to Page 16, line 16 of the originally filed application.
[70] See element 12 in Figs. 1 and 3, and Page 12, line 29 to Page 13, line 19.
[71] See elements 12 and 14 in Figs. 1 and 2, Page 12, line 14 to Page 13, line 19.

on, for example, the master sound masking units 14, the In-room switch 24, and the control unit 12.[72]

Claim 116

Claim 116 is an independent claim directed to a networkable sound masking device. The device of claim 116 includes an interface for interfacing to a network. This reads on, for example, communication interface 55.[73] The device of claim 116 further includes a processor configured for receiving one or more control signals from the interface, the one or more control signals being intended for the networkable sound masking device and the one or more control signals comprising a masking volume signal and a masking frequency signal. This reads on, for example, the digital signal processing module 50 and the control unit 12.[74] The processor is configured to generate a sound masking signal. This reads on, for example, the digital signal processing module 50 and the response of the processing module 50 to the control unit 12.[75] The device of claim 116 further includes an output stage configured to output the sound masking signal. This reads on, for example, audio power amplifier stage 52.[76]

Claims 117-119

Claim 117 is dependent from claim 116, and includes all the elements of claim 117. Claim 117 further defines the interface as including an address component configured to recognize the one or more control signals intended for the networkable sound masking device. This reads on, for example, the combination of the digital signal processing module 50 and executable code or firmware.[77]

Claim 118 is dependent from claim 116, and includes all the elements of claim 116. Claim 118 further defines the output stage in the networkable sound masking device as comprising an amplifier and the processor being configured to control the

[72] See elements 12, 14 and 24, Page 14 to Page 13, line 19 and Page 28, line 7 to Page 29, line 3 of the originally filed application.
[73] See element 55 in Fig. 2, and Page 15, line 27 to Page 16, line 4 of the originally filed application.
[74] See element 50 in Fig. 2 and Page 14, lines 10-19 of the originally filed application.
[75] See element 50 in Fig. 2 and Page 14, lines 10-19, and element 12 and Page 12, line 29 to Page 13, line 19 of the originally filed application.
[76] See element 52 in Fig. 2, and Page 15, lines 15-24 of the originally filed application.
[77] See elements 50, 55 in Fig. 2, and Page 15, line 27 to Page 16, line 16.

output stage. This reads on, for example, audio power amplifier stage 52 and digital signal processing module 50 and executable code or firmware.[78]

Claim 119 is dependent from claim 116, and includes all the elements of claim 116. Claim 118 further defines the processor as including a random noise generator have an output coupled to an equalizer stage. This reads on, for example, the random noise generator 66 and the equalizer 68.[79]

VI. 41.37(c)(1)(vi) GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

The Appellant seeks the Board's review of the following rejections:

A. Claims 108-118 stand finally rejected under U.S.C. § 103(a) as being unpatentable over U.S. Patent No. 4,319,088 ("Orfield") in view of U.S. Patent Application Publication No. 2002/0072816 ("Shdema").[80]

B. Claim 119 stands finally rejected under U.S.C. § 103(a) as being unpatentable over U.S. Patent No. 4,319,088 ("Orfield") as modified by U.S. Patent Application Publication No. 2002/0072816 ("Shdema") as applied to claim 116, and in further view of U.S. Patent No. 4,686,693 ("Ritter").[81]

Claims 108-119 are being appealed.

VII. (c)(1)(vii) ARGUMENT

A. Claims 108-118 are not rendered obvious by Orfield in view of Shdema

The Appellant requests the Board reverse the Examiner's rejection of claims 108-118 under U.S.C. § 103(a) as allegedly being obvious over Orfield in view of Shdema. In support of the Appellant's request, the following arguments are submitted.

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[78] See elements 50, 55 in Fig. 2, and Page 15, line 27 to Page 16, line 16 of the originally filed application.
[79] See elements 66, 68 in Fig. 2, and Page 14, line 20 to Page 15, 9 of the originally filed application.
[80] See page 2 of the August 24, 2009 Final Office Action.
[81] See page 10 of the August 24, 2009 Final Office Action.

The Examiner contends that Orfield teaches a communication network, a plurality of sound masking units, each having a sound masking signal generator, and a control unit configured to generate one or more control signals including the masking volume and the masking frequency signal, as recited in independent claims 108 and 114.

The Examiner relies on Fig. 3 of Orfield, specifically, the detachable cable 18 connecting the master sound masking unit 14 to the slave sound masking unit 16, as teaching a communication network. At Col. 4, lines 22-23 and lines 28-35, Orfield describes the connection between the master sound masking unit 14 and the slave sound masking unit 16 as follows:

> In order to connect master unit 14 to slave unit 16, an interconnect cable 18 is employed.
>
> A flexible cable 18 includes conductors necessary to carry the signal from the master unit tapped at point 64 on Fig. 1 to the slave unit 16 at point 86 on Fig. 2. Conductors may also be provided within cable 18 for carrying an alternative sound source signal which may be inputted to voice coil 68 of the slave unit 16.

With reference to the above-noted passages, "point 64" on Fig. 1 is described at Col. 4, lines 5-8 as follows:

> The slave output 64 from the octave equalizer 24 is connected to speaker 62 via voice coil 66. Alternative sound sources may also be inputted to speaker 62 via the second voice coil 68.

At Col. 5, line 56 to Col. 6, line 8, Orfield further describes the function of the cable 18 as follows:

> A cable 18 is connected at jack 70 of the master unit and run to one of the slave units 16, and is connected at jack 71 of the slave unit. FIG. 4 illustrates a grid layout having three zones determined by the above analysis to have similar ambient noise characteristics, indicated by the area encompassed by the three separate chains of sound masking units. Each chain contains a master unit 14 and a plurality of slaves 16 (shown as single black dots). As mentioned above, master unit 14 contains two jacks 70 which permit two strings of slave units to be attached at the master. This is the preferable connection in order to reduce line losses. Each slave unit 16 has a jack 71 and jack 70 (not shown). One of

> the jacks 71 in each slave 16 carries the signal from the master 14 to respective voice coil 62 and the jack 70 in the slave passes the signal on further slaves 16 via cable 18. Cable 18 carries a masking sound output signal from the master where it is generated. Also carried within cable 18 may be the alternative sound signals which are input to the master at jack 92.

With all due respect, it is submitted that this is not a communication network, nor is it a communication network that provides control information or control signals. Orfield is limited to teaching a cable connection between the master sound masking unit 14 and one or two slave units 16. The cable is for outputting the sound masking signal generated by the octave equalizer 24 (Fig. 1) in the master sound masking unit 14 to the respective connected slave units 16, as clearly shown in Fig. 1.

This deficiency in Orfield is further borne out by the fact that the "SLAVE OUT" from the master sound masking unit 14 is directly connected to point 86 (i.e. "FROM MASTER) of the slave sound masking unit 16 as shown in Fig. 2. As shown in Fig. 2, point 86 is electrically connected to a potentiometer 76 which is in turn connected to a voice coil 66. The voice coil 66 is coupled to a speaker 62, as also shown in Fig. 2. According to Orfield, the slave unit 16 can include a second voice coil 68 for receiving a signal from an alternate sound source control module 14 and coupling the alternative signal to the speaker 62. The speakers 26 and 62 as taught by Orfield comprise a conventional passive speaker that emits the sound masking signal generated by the octave equalizer 24 in the master sound masking unit 14, and applied to the slave sound masking unit 16 via the cable 18. Accordingly, the cable 18 as taught by Orfield does not comprise a communication network, but is merely, a wire connection for passing the sound masking signal generated by the master unit 14 to the slave unit 16 for emission on the speaker 62.

Furthermore, Orfield does not describe or teach any form of control message, control signal or control command transmitted over the cable 18. Nor does Orfield disclose, teach or suggest that the speakers 26 and/or 62 have any form of capability to receive a masking volume control signal or a masking frequency control signal as recited in independent claims 108, 114 and 116. If the cable 18 was utilized as a communication network as alleged, and control signals were somehow imposed on the cable 18, then according to the circuitry disclosed and taught by Orfield, the control

signals, e.g. the masking frequency control signal and/or the masking volume control signal, would be played over the speakers 26 and 62, and since the control signals do not comprise a sound masking signal or a paging signal, the result would be an unintended audio signal that in essence would destroy the sound masking functionality of the system as disclosed and taught by Orfield. In view of the foregoing, it is submitted that the cable 18 does not constitute a communication network as alleged, and the limitation in claims 108 and 114 are not met.

The non-networked configuration of the system disclosed and taught by Orfield is further demonstrated by the arrangement of multiple sound masking systems shown in Figs. 3 and 4. While Orfield discloses a multiple sound masking unit configuration, Orfield does not teach, or even suggest, control of the plurality of sound masking units, nor does Orfield teach multiple sound masking units that are interconnected in a communication or control network. For example, at Col. 5, lines 6-12, Orfield describes the control function as follows:

> Since only the master units 14 are capable of frequency control and since only one master is used per zone, the zones are defined by areas of equal or nearly equal ambient noise characteristics. Fig. 4 illustrates an arrangement of 3 distinct zones each having a master unit 14 and slave units (unnumbered) designated by black dots.

As described in more detail below, any control function according to Orfield is limited to the teaching of a manual adjustment of the potentiometers 44 and 48. As such, Orfield falls well short of disclosing, teaching or suggesting a communication network.

The Examiner relies on Fig. 1 of Orfield, specifically, the amplifier 22 and the potentiometer 48 as teaching the masking volume control signal, and the octave equalizer 24 and the potentiometer 46 as teaching the masking frequency control signal. At Col. 3, lines 56-62 and line 67 to Col. 4, line 2, Orfield describes the potentiometers as follows:

> Master unit 14 also includes potentiometers 42, 44, 46 and 48, which are mounted on face 38 of enclosure 19. Unit 14 is preferably mounted on rubber feet 51 on top of ceiling tile 52 as shown in Fig. 5, with holes drilled in tile 52 to permit potentiometers 42-48 to extend therethrough for easy adjustment

> without the need to open the ceiling to make certain adjustments....Potentiometer 48 is a frequency trimming control also known as a tone control, which adjusts the frequency output of amplifier.

According to Orfield, the potentiometer 44 is manually adjusted to control the signal strength fed into the voice coil 30 of speaker 26. With all due respect, the manual adjustment of the potentiometer 44 is simply changing the resistance value of the potentiometer 44 and does not meet the limitation of the masking volume signal as recited in claims 108 and 114. Furthermore, even if the adjustment of the potentiometer 44 could be construed as generating a masking volume signal (which is not being conceded), the masking volume signal so generated is not received over the communication network as further recited in claims 108 and 114. According to Orfield, the cable 18 between the master unit 14 and the slave unit 16 is configured for passing the sound masking signal output from IO Octave Equalizer 24 as clearly shown in the circuit schematic of Fig. 1. Similarly, the potentiometer 48 is manually adjusted to control the frequency or tone output of the amplifier 22. Again and with all due respect, the manual or hand adjustment of the potentiometer 48 is simply changing the resistance value of the potentiometer 48 and as such does not meet the limitation of the masking frequency control as recited in claims 108 and 114. Furthermore, even if the adjustment of the potentiometer 48 could be construed as generating a masking frequency signal (which is not being conceded), the signal so generated is applied directly to the amplifier 22 (as shown in Fig. 1) and not received over the communication network as further recited in claims 108 and 114. As shown in Fig. 1 of Orfield, the cable 18 between the master unit 14 and the slave unit 16 is configured for passing the sound masking signal output from IO Octave Equalizer 24, and the output of the potentiometer 48 is electrically isolated from the cable 18 as also clearly shown in the circuit schematic of Fig. 1.

It is further noted that the Examiner appears at Page 3 of the Final Action to have characterized the master sound masking unit 14 according to Orfield as corresponding to both the "sound masking unit" recited in the second clause of claim 108 and the "control unit" recited in the third clause of claim 108. In addition to the deficiencies identified above, this characterization gives rise to a further inconsistency as the master sound masking unit 14 Orfield cannot function as both a sound

masking unit and as a control unit as recited in claim 108. Further, even if the characterization is accepted that the master sound masking unit 14 according to Orfield is a "control unit" *per se* (which is not being conceded), the slave sound masking units 16 coupled by the interconnect cable 18 would need to be the plurality of sound masking units (which is not being conceded), and since the slave sound masking units 16 as taught by Orfield do not include any form of a sound masking generator, but are merely limited to a speaker 68 as shown in Fig. 2, the limitations of the second clause of claim 108 are still not met. Appellants assert this argument equally applies to claim 114.

With regard to the control unit recited in the third clause of claim 108, the Examiner alleges at Page 3 of the Office Action, that master sound masking unit 14 taught by Orfield meets the limitations of the control unit. As discussed above, if the master sound masking unit 14 is characterized as the "control unit", then the slave sound masking unit 16 must be characterized as the "sound masking unit". Notwithstanding the apparent inconsistency between characterizing the master sound masking unit 14 as both a sound masking unit and a control unit as discussed above, it is submitted that even if the master sound masking unit 14 is characterized as a "control unit", the limitations of claim 108 are not met. First, the master sound masking unit 14 does not generate a masking volume control signal to selectively control operation of the plurality of sound masking units, i.e. the slave sound masking unit 16. The potentiometer 44 is manually adjusted to vary the resistance of the potentiometer and thereby vary the amplitude of the output signal applied to the speaker 26 as clearly shown in Fig. 1 of Orfield. The manual adjustment of the potentiometer 44 does not amount to a masking volume control signal, nor does the potentiometer 44 generate a masking volume control signal, and nor can the output of the potentiometer 44 be transmitted to the slave sound masking unit 16, as clearly apparent from the circuit schematic in Fig. 1 of Orfield. Second, the master sound masking unit 14 as taught and described by Orfield does not generate a masking frequency control signal to selectively control operation of the plurality of sound masking units (i.e. the slave sound masking unit 16) as recited in claim 108. As taught and disclosed by Orfield, the potentiometer 48 is manually adjusted to vary the resistance of the potentiometer and thereby vary or trim the frequency response of the

amplifier 22 as clearly shown in Fig. 1 of Orfield. The manual adjustment of the potentiometer 48 does not amount to a masking volume control signal, nor does the potentiometer 48 generate a masking volume control signal to selectively control the operation of the slave sound masking unit 16, and nor can the output of the potentiometer 48 be transmitted to the slave sound masking unit 16. The potentiometer 48 is electrically isolated from the output to the slave masking unit 16 as clearly shown in the circuit schematic in Fig. 1 of Orfield. Third, the slave sound masking unit 16 as disclosed and taught by Orfield is not configured with a sound masking generator as recited in the second clause of claim 108, and therefore the operation of the slave sound masking unit 16 as taught by Orfield cannot be selectively controlled by the masking volume and masking frequency signals as recited in the third clause of claim 108. Fourth, Orfield actually teaches away from the recited configuration of the masking volume and masking frequency signals by disclosing and teaching the manual adjustment of the potentiometers. For instance, Orfield teaches manually operated shafts for the potentiometers 44 and 48, as shown in Fig. 6 and described at Col. 5, lines 46-50. In view of these deficiencies, the master sound masking unit 14 as taught by Orfield does not meet the limitations of the control unit as recited in the third clause of claim 108.

In view of the foregoing, it is submitted that Orfield does not meet at least the following limitations as recited in independent claim 108:

- a communication network
- a plurality of sound masking units including a communication interface
- a sound masking signal generator responsive to a masking volume signal
- a sound masking signal generator responsive to a masking frequency signal
- a control unit having a communication interface
- a control unit configured to generate the masking volume signal and the masking frequency signal

Appellants respectfully submit these arguments similarly apply to claim 114.

The Examiner acknowledges at Page 3 that Orfield does not disclose "digital processor receiving and transmitting control signals over said communication network; and a control unit, said control unit having a communication interface for coupling to said communication network for transmitting said control signals to selectively control operation of said plurality of sound masking units". The Applicant agrees with the Examiner's recognition of these deficiencies and asserts they apply to independent claims 108, 114 and 116.

The Examiner relies on Shdema as teaching a digital processor receiving and transmitting control signals over the communication network (Figs. 1-5) and an audio speaker system network comprising a plurality of speaker units including a communication interface for coupling the speaker units (114) (i.e. sound masking units) to the communication network for receiving and transmitting control signals over the communication network (Fig. 1; Page 3, [0028]-[0030])

Based on these characterizations, the Examiner contends that Shdema teaches the deficiencies of Orfield with respect to the invention as defined by independent claims 108, 114 and 116, and that therefore one skilled in the art would have applied these teachings to Orfield.

It is respectfully submitted that there is no motivation for one skilled in the art to combine the references for the reasons as discussed below. Secondly, even if one skilled in the art were to combine the teachings of Orfield and Shdema as suggested by the Examiner, Shdema does not remedy the deficiencies of Orfield and the resulting system is not the same as that defined by claims 108, 114 and 116.

First, it is to be appreciated that audio and music systems, such as taught by Shdema, and sounds masking systems as taught by Orfield are fundamentally different systems directed to solve fundamentally different problems. The Shdema system is concerned with delivering intelligible audio throughout a house. Sound masking systems, on the other hand, are concerned with suppressing, i.e. masking, unwanted sounds or ambient sounds in a physical space such as an office or workplace. Sound masking systems generate incoherent or unintelligible background

sounds that serve to mask the unwanted intelligible sounds in the workplace. Because ambient sounds can vary from location to location in a workplace, the space may be divided into one or more zones, with each zone having a sound masking signal with a different masking level and/or frequency level, wherein the masking level or frequency level is tailored to the ambient sounds sought to be masked or suppressed.

It is also to be appreciated that audio and music systems, such as taught by Shdema, and sound masking systems, such as taught by Orfield, are fundamentally different systems directed to solve fundamentally different problems. The Shdema system is concerned with delivering intelligible audio throughout a house. Sound masking systems, on the other hand, are concerned with suppressing, i.e. masking, unwanted sounds or ambient sounds in a physical space, typically an office or workplace, and as such are designed to adjust different acoustical parameters from intelligible music or audio systems. Sound masking systems generate incoherent or unintelligible background sounds that serve to mask the unwanted intelligible sounds in the workplace. Because ambient sounds can vary from location to location in a workplace, the space may be divided into one or more zones, with each zone having a sound masking signal with a different masking level and/or frequency level, wherein the masking level or frequency level is tailored to the ambient sounds sought to be masked or suppressed.

In this regard, reference is made to the state of art, for example, as described in US Patent No. 4,185,167 to Cunningham which was cited as prior art in a previous Office Action by the Examiner. At Col. 1, lines 14 to 23, Cunningham states as follows:

> Such proposals have included ... the use of piped-in or canned music in an attempt to condition the environment to reject the unwanted sounds in the area occupied by the listener. However, music itself played continuously may become distracting to the listener or listeners over an extended period of time, particularly if the music is of a type which the listener may not find pleasing.

Cunningham clearly distinguishes between intelligible sounds, such as music, and unintelligible or incoherent sounds for masking unwanted sounds. Cunningham further emphasizes that intelligible sounds, such as music, are not suitable for sound masking and can disturb occupants over time. Cunningham clearly teaches away from

the use of intelligible sounds, e.g. music, in sound masking systems. Shdema, on the other hand, discloses and teaches an audio system for operating a plurality of speakers comprising an audio management system, a user interface and an audio source cluster configured for the distribution of intelligible audio signals from a central source (i.e. the audio source cluster) to the plurality of speakers. As taught by the prior art (for example, Cunningham) and clearly within the knowledge of one skilled in the art, such audio signals are not suitable for sound masking. As a result, one skilled in the art would not be led or motivated by the art to combine the teachings of Shdema with Orfield as alleged. Furthermore, Shdema does not disclose or teach utilizing a sound masking signal in the "audio source cluster" (106 in Figs. 1 and 3) as an input to the audio management system (102 in Figs. 1 and 2, or 1100 in Fig. 11), nor does Shdema disclose or teach generating control signals such as a masking volume signal and/or a masking frequency signal, nor does Shdema teach the speakers including a sound masking generator or generating a sound masking signal, nor does Shdema disclose or teach controlling a plurality of sound masking units in a network.

It is further submitted that even if one skilled in art were to combine the teachings of Orfield and Shdema (notwithstanding the lack of any motivation or suggestion in the art) as alleged by the Examiner, Shdema does not remedy the deficiencies of Orfield. In particular, Shdema does not disclose or teach a control unit configured to generate control signals such as a masking volume signal or a masking frequency signal. Furthermore, Shdema does not disclose or teach a control unit having a communication interface for coupling to a communication network for transmitting the masking volume signal and/or the masking frequency signal. Accordingly, if one skilled in the art were to combine Orfield and Shdema as alleged by the Examiner, the resulting system would still not include the capability to generate a sound masking volume signal and a sound masking frequency signal, which are transmitted over a network to control the volume and/or frequency of a sound masking signal generated locally at a sound masking unit coupled to the network. In other words, both Orfield and Shdema at least lack the teaching or suggesting of generating a masking volume signal or a masking frequency signal and providing a

control unit to transmit these control signals over a communication network to selectively control operation of the sound masking units.

The novelty and inventive ingenuity of the present invention is further borne out by the fact that in the prosecution of the subject application at least 4 prior art references were cited in the field of sound masking art, but not a single one of the cited references taught, disclosed or even suggested configuring sound masking units in a communication network and/or utilizing a control unit to generate control signals and transmit the control signals over the communication network to selectively control operation of the sound masking units. In view of the crowded nature in the sound masking art, it is respectfully submitted that if such a solution was obvious, one skilled in the art would have considered such a solution prior to the present invention.

Since Orfield and Shdema, whether taken alone or in combination, do not disclose or teach all of the limitations as recited in independent claims 108, 114 and 116, it is submitted that the invention as recited is not obvious. Since the remaining claims depend either directly or indirectly from the associated independent claim, it is submitted that the dependent claims are also not obvious for similar reasons. Accordingly, the Appellant respectfully requests that the Examiner's rejection of claim 108-118 under U.S.C. § 103(a) as allegedly being obvious over Orfield in view of Shdema be reversed.

B. Claim 119 is not obvious over Orfield in view of Shdema and further in view of Ritter

The Appellant requests the Board reverse the Examiner's rejection of claim 119 under U.S.C. § 103(a) as allegedly being obvious over Orfield in view of Shdema, and in further view of Ritter. In support of the Appellant's request, the following arguments are submitted.

Claim 119 is dependent on independent claim 116. Since Orfield and Shdema, whether taken alone or in combination, do not disclose or teach all of the limitations as recited in independent claim 116, for the reasons as discussed above, it is

submitted that associated dependent claim 119 is also obvious. Accordingly, the Appellant respectfully requests that the Examiner's rejection of claim 119 under U.S.C. § 103(a) as allegedly being obvious over Orfield in view of Shdema, and in further in view of Ritter be reversed.

VIII. CONCLUSION

In view of the detailed discussion provided above regarding the pending rejections, the Appellant respectfully submits that the bases for the rejections have been addressed and overcome, leaving the present application in condition for allowance. Therefore, the Appellant respectfully requests the Board to reverse the Examiner's rejection of claims 108 to 119.

Respectfully submitted,

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41.37(c)(1)(viii) – Claims Appendix

1-107. Cancelled.

108. (Previously Presented) A sound masking system for masking sound in a physical environment, said sound masking system comprising:

a communication network for said physical environment;

a plurality of sound masking units, at least some of said sound masking units including a digital processor configured for a sound masking signal generator and a communication interface for coupling to said communication network for receiving a plurality of control signals over said communication network including a masking volume signal and a masking frequency signal, and said sound masking signal generator being responsive to said masking volume signal and said sound masking frequency signal for generating a sound masking output signal, said sound masking output signal having a volume derived from said masking volume signal and a frequency characteristic derived from said sound masking frequency signal;

a control unit configured to generate said control signals including said masking volume signal and said masking frequency signal, and said control unit having a communication interface for coupling to said communication network for transmitting said control signals to selectively control operation of said plurality of sound masking units.

109. (Previously Presented) The sound masking system as claimed in claim 108, wherein said sound masking unit includes an address component for recognizing control signals intended for the sound masking unit associated with said address component.

110. (Previously Presented) The sound masking system as claimed in claim 108, wherein said plurality of sound masking units are associated with a plurality of sound masking zones, each of said sound masking units being associated with one of said plurality of sound masking zones, and said sound masking units providing sound masking for said associated sound masking zone independently of said other sound masking zones.

111. (Previously Presented) The sound masking system as claimed in claim 110, wherein said sound masking units associated with each of said sound masking zones are configured to provide a sound masking output tailored for said associated sound masking zone and said sound masking output being based on said masking volume and said masking frequency signals.

112. (Previously Presented) The sound masking system as claimed in claim 108, further comprising a plurality of zones, and one or more of said sound masking units being configured for one or more of said zones.

113. (Previously Presented) The sound masking system as claimed in claim 112, wherein said zones includes one or more of a sound masking zone, a timer zone, and a switch zone.

114. (Previously Presented) A sound masking system for controlling the ambient noise in a physical environment, said sound masking system comprising:

a communication network for said physical environment;

a plurality of sound masking units, at least some of said sound masking units including a sound masking generator comprising a processor configured to generate a sound masking signal and a communication interface for coupling to said communication network for receiving one or more control signals over said communication network including a masking volume signal and a masking frequency signal, and said sound masking generator being responsive to said masking volume signal and said sound masking frequency signal for generating said sound masking signal;

a control unit configured to generate said one or more control signals including said masking volume signal and said masking frequency signal, and said control unit having a communication interface for coupling to said communication network for transmitting said one or more control signals to selectively control operation of said plurality of sound masking units;

a plurality of zones, and one or more of said sound masking units being configured for one or more of said plurality of zones.

115. (Previously Presented) The sound masking system as claimed in claim 114, wherein said zones include one or more of a sound masking zone, a non-masking zone, a timer zone, and a switch zone.

116. (Previously Presented) A networkable sound masking device comprising:

an interface configured to interface to a network;

a processor configured to receive one or more control signals from said interface, said one or more control signals being intended for the networkable sound masking device and said one or more control signals comprising a masking volume signal and a masking frequency signal;

said processor being configured to generate a sound masking signal in response to said masking frequency signal; and

an output stage configured to output said sound masking signal.

117. (Previously Presented) The networkable sound masking device as claimed in claim 116, wherein said interface includes an address component configured to recognize said one or more control signals intended for the networkable sound masking device.

118. (Previously Presented) The networkable sound masking device as claimed in claim 116, wherein said output stage comprises an amplifier and said processor being configured to control said output stage in response to said masking volume signal.

119. (Previously Presented) The networkable sound masking device as claimed in claim 116, wherein said sound masking module comprises a random noise generator having an output coupled to an equalizer stage, and said processor being configured to control said equalizer stage in response to said masking frequency signal.

41.37(c)(1)(ix) – Evidence Appendix

None.

41.37(c)(1)(x) - Related Proceedings Appendix

None.